MultiVoice Gateway Configuration
In-call DTMF detection for IPDC

## IPDC messages that support in-call DTMF detection

Following are IPDC messages (as defined in IPDC specification *Level 3 Communications, Internet Protocol Device Control (IPDC), Revision 0.15)* that support in-call DTMF dection and notification.

*Table 2-19. IPDC messages supporting in-call DTMF detection and generation*

IPDC message	Tag	Tag values
RCCP	0x75 (Constant DTMF Tone Detection)	• 0x00 - DTMF tone detection off • 0x01 - DTMF tone detection on If a tag is missing, DTMF tone detection is off.
RMCP	0x75 (Constant DTMF Tone Detection)	• 0x00 - DTMF tone detection off • 0x01 - DTMF tone detection on If a tag is missing, there is no affect on the current state of detection.
NTN	0x49 (Tone Type)	• 0x01 - DTMF tone
NTN	0x33 (Tone String)	This tag contains the DTMF digit that was detected. The length of this tag value is 1.

Call Flow

In the following call flow, a packet call is setup with DTMF detection enabled. After two DTMF digits are entered, the call is modified to disable DTMF detection.

```
      TAOS Unit                     TAOS Unit
      (incoming)     Softswitch     (outgoing)
      ----------     ----------     ----------
          |              |              |
          |              |     RCCP     |
          |              |------------->|
          |              |     ACCP     |
          |              |<-------------|
          |     RCCP     |              |
          |<-------------|              |     Tag 0x75 = 0x01
          |     ACCP     |              |
          |------------->|              |
          |     RTP      |              |
          |<------------------------------>| Voice exchange in
            progress
```

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```
                    |            |           |
user enters         |    NTN     |           |
a dtmf digit        |----------->|           | Tag 0x33 = <digit>
                    |            |           |
user enters         |    NTN     |           |
a dtmf digit        |----------->|           | Tag 0x33 = <digit>
                    |            |           |
                    |    RMCP    |           |
                    |<-----------|           | Tag 0x75 = 0x00
                    |    AMCP     |           |
                    |----------->|           |
user enters         |            |           |
a dtmf digit        |            |           |
                    |            |           |
user enters         |            |           |
a dtmf digit        |            |           |
                    |            |           |
```
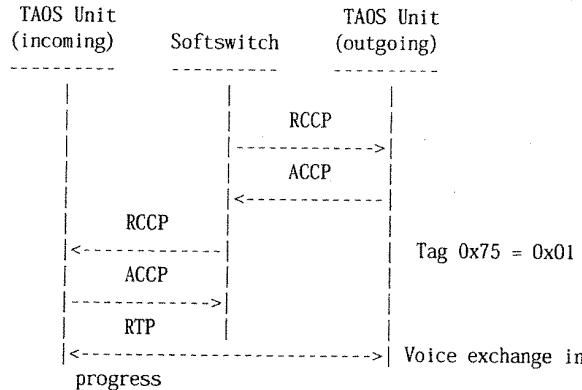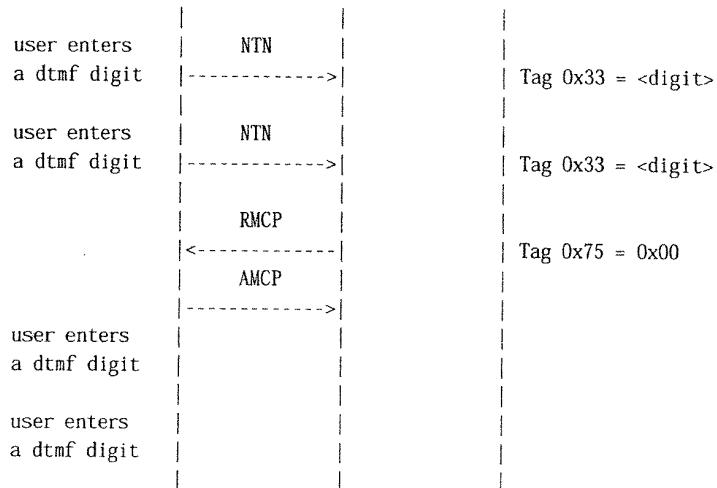
Interaction with break-in voice announcements

If in-call DTMF detection is enabled and a break-in announcement is played (see "Break-in voice announcements in IPDC" on page 4-6 for details), the first DTMF entered will stop the announcement:

```
TAOS Unit                      TAOS Unit
(incoming)     Softswitch      (outgoing)
----------    ----------      ----------
     |            |      RCCP     |
     |            |-------------->|
     |            |      ACCP     |
     |            |<--------------|
     |    RCCP    |               |
     |<-----------|               | Tag 0x75 = 0x01
     |    ACCP    |               |
     |----------->|               |
     |    RTP     |               |
     |<----------------------------->| Voice exchange in
     |            |               |   progress
     |            |               |
     |    STN     |               |
     |<-----------|               |
     |    ASTN    |               |
     |----------->|               |Announcement starts
     |            |               |
user enters |    NTN     |        |
a dtmf digit|----------->|        |Tag 0x33 = <digit>
     |    ASTN    |               |
     |----------->|               |Announcement stops
```
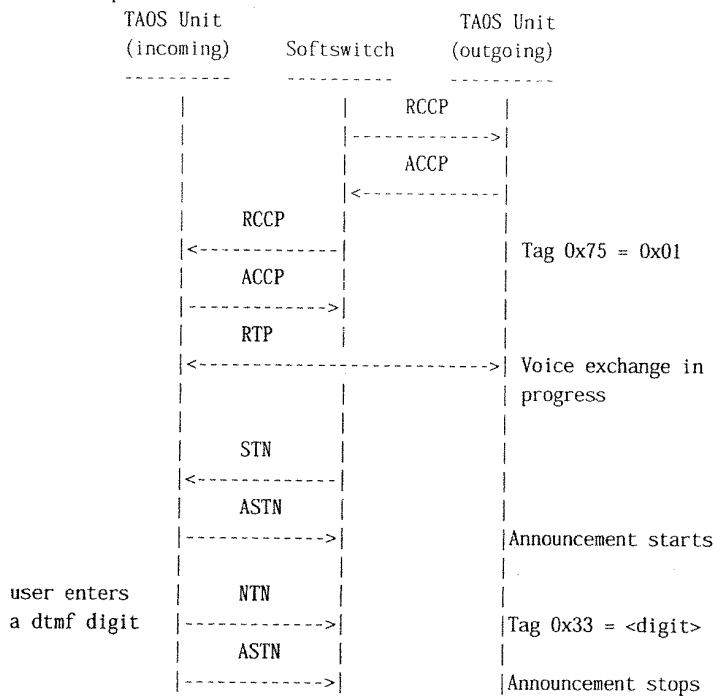
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An RMCP that is received by the MultiVoice Gateway while a break-in announcement is playing is rejected. An MRJ will be sent with Tag 0xFE (Cause Code) set to 0x65 (Message Not Compatible With Call State).

Note If DTMF is being carried inband, then the first DTMF digit entered during a break-in announcement is not played out to the other party.

Call re-origination

In-call DTMF detection can be combined with existing IPDC support on the MultiVoice Gateway to provide a call re-origination application.

Using in-call DTMF detection, the MultiVoice Gateway forwards DTMF received during an active packet call to the Softswitch. The DTMF is sent in the NTN message, one digit per message. The Softswitch monitors the received DTMF stream for a pattern (for example, **9) that indicates that the calling party wishes to terminate the active call and start a new call.

The Softswitch then sends an RCR, waits for the ACR, then sends an LTN to start the two-stage dialing for the next call, while maintaining the signaling for the incoming CIC with the PSTN.

The RCR tells the incoming MultiVoice Gateway to terminate the VoIP call. This tears down the VoIP call route and frees the resources associated with the VoIP call. The ensuing LTN would be identified as the first one for a call. This tells the incoming MultiVoice Gateway to setup a new VoIP call route.

The following call flow shows how in-call DTMF detection is utilized for call re-origination.

```
SS7 Network     TAOS Unit                    TAOS Unit       SS7 Network
(incoming)      (incoming)     Softswitch    (outgoing)      (outgoing)
- - - - - - -   - - - - - -    - - - - - -   - - - - - -     - - - - - - -
    |    IAM       |                             |               |
    |- - - - - - - - - - - - - - - ->|           |               |
    |              |    LTN          |           |               |
    |              |<- - - - - - - - |           |               |
    |              |    ALTN         |           |               |
    |              |- - - - - - - - ->|          |               |
    |    ACM       |                 |           |               |
    |<- - - - - - - - - - - - - - - -|           |               |
    |    ANM       |                 |           |               |
    |<- - - - - - - - - - - - - - - -|           |               |
A:  |              |    STN          |           |               |
    play          |<- - - - - - - - |           |               |
    announcement  |    ASTN         |           |               |
    to enter      |- - - - - - - - ->|          |               |
    dnis          |    ASTN         |           |               |
                  |- - - - - - - - ->|          |               |
                  |    ALTN         |           |               |
    dnis entered  |- - - - - - - - ->|          |               |
                  |                 |           |    IAM         |
                  |                 |- - - - - - - - - - - - - - ->|
                  |                 |           |    ACM         |
                  |                 |<- - - - - - - - - - - - - - -|
```

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```
                                            |   ANM    |
                                        |<------------------------|
                                        |   RCCP   |
                                        |------------>|
                                        |   ACCP   |
                                        |<------------|
                               RCCP     |
packet call          |<-------------|
set up                   ACCP     |
dtmf detection |------------>|
enabled                  RTP      |
                               |<----------------------->|

user enters            NTN      |
a dtmf digit      |------------>|

user enters            NTN      |
a dtmf digit      |------------>|

softswitch recognizes           |
the 2 digits as the             |
pattern to initiate             |
the next call                   |

softswitch              |        |   REL    |
releases other   |------------------------------->|
end (both               |   RCR    |
packet and       |        |------------>|
circuit legs            |   ACR    |
of call)                |        |<------------|
                        |        |   RLC    |
                        |        |<----------------------|

                               RCR      |
release          |<-------------|
packet call           ACR      |
                  |------------>|

                               LTN      |
start a new      |<-------------|
2-stage call          ALTN     |
                  |------------>|

(goto "A:"
above)
```

Note Call re-origination when signaled over SS7 VoIP does not use the voip profile parameters sequential-call-enable and next-call. These parameters are used when call re-origination is signaled over H.323 VoIP.

End-of-call break-in voice announcements

This section illustrates call flows for the special case of playing a break-in voice announcement at the end of a call. Such an announcement could be played either before or after the call is released, with VoIP call persistence enabled or disabled.

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In-call DTMF detection for IPDC

• End-of-call break-in announcement is played after call release, with VoIP call
  persistence mode disabled.

```
    TAOS Unit                            TAOS Unit
    (incoming)        Softswitch         (outgoing)
   - - - - - - - -   - - - - - - - -    - - - - - - -
        |                 |                  |
        |       RTP       |                  |
        |<- - - - - - - - - - - - - - - - - - >|
        |                 |       RCR        |
        |                 |- - - - - - - - - >|
        |                 |       ACR        |
        |                 |<- - - - - - - - - |
        |       RCR       |                  |
        |<- - - - - - - - |                  |
        |       ACR       |                  |
        |- - - - - - - - >|                  |
        |       STN       |                  |
        |<- - - - - - - - |                  |
        |      ASTN       |                  |
        |- - - - - - - - >|                  |
        |      ASTN       |                  |
        |- - - - - - - - >|                  |
```
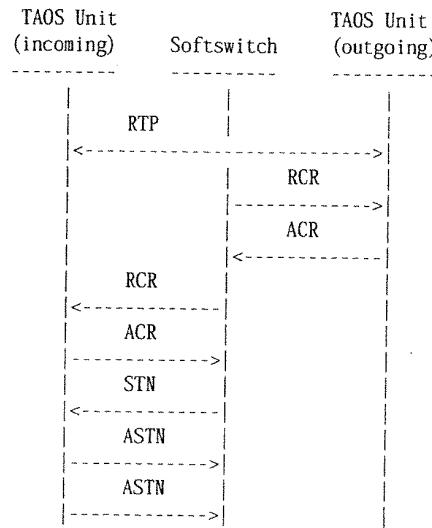
The STN results in the setup of a new VoIP call route. Note that this messaging is
possible without support for break-in announcements by using existing capabilities.

• End-of-call break-in is played after call release, with VoIP call persistence mode
  enabled.
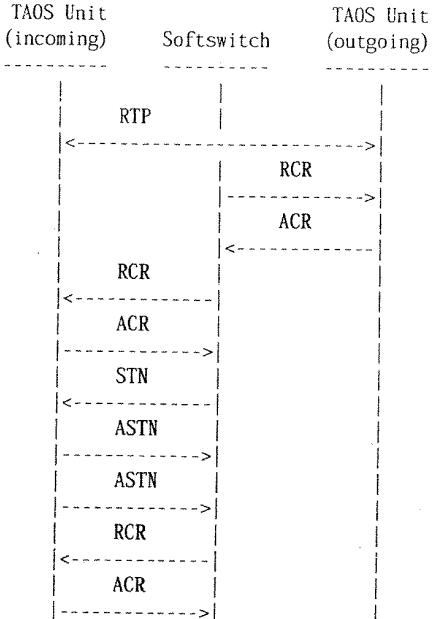
```
    TAOS Unit                            TAOS Unit
    (incoming)        Softswitch         (outgoing)
   - - - - - - - -   - - - - - - - -    - - - - - - -
        |                 |                  |
        |       RTP       |                  |
        |<- - - - - - - - - - - - - - - - - - >|
        |                 |       RCR        |
        |                 |- - - - - - - - - >|
        |                 |       ACR        |
        |                 |<- - - - - - - - - |
        |       RCR       |                  |
        |<- - - - - - - - |                  |
        |       ACR       |                  |
        |- - - - - - - - >|                  |
        |       STN       |                  |
        |<- - - - - - - - |                  |
        |      ASTN       |                  |
        |- - - - - - - - >|                  |
        |      ASTN       |                  |
        |- - - - - - - - >|                  |
        |       RCR       |                  |
        |<- - - - - - - - |                  |
        |       ACR       |                  |
        |- - - - - - - - >|                  |
```

The STN results in the setup of a new VoIP call route. A second RCR is required to destroy the VoIP call route setup by the STN to play the break-in announcement.

Note The exception to this is if call re-origination is in progress. In this case, the second RCR is replaced with a LTN/STN/RCCP signaling at the start of the next call. The VoIP call route that was set up for the break-in announcement is then re-used.

- End-of-call break-in is played before call release, VoIP call persistence mode enabled or disabled.

```
         TAOS Unit                        TAOS Unit
        (incoming)      Softswitch        (outgoing)
        - - - - - - - -    - - - - - - - -    - - - - - - - -
            |                  |                  |
            |       RTP        |                  |
            |<- - - - - - - - - - - - - - - - - ->|
            |                  |       RCR        |
            |                  |- - - - - - - - - ->|
            |                  |       ACR        |
            |                  |<- - - - - - - - - -|
            |       STN        |                  |
            |<- - - - - - - - - |                  |
            |       ASTN       |                  |
            |- - - - - - - - - ->|                  |
            |       ASTN       |                  |
            |- - - - - - - - - ->|                  |
            |       RCR        |                  |
            |<- - - - - - - - - |                  |
            |       ACR        |                  |
            |- - - - - - - - - ->|                  |
```
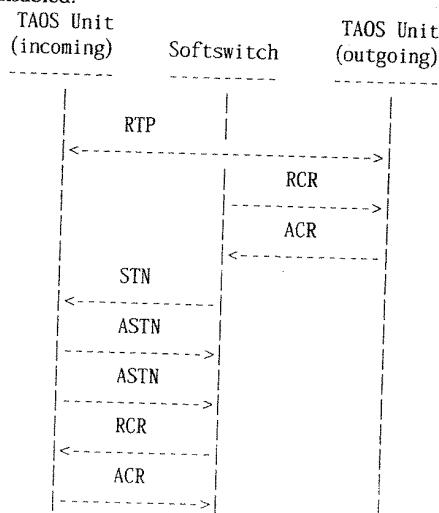
This messaging is possible with support for break-in announcements. The STN utilizes the existing VoIP call route for the call. This reduces gateway processing, but adds extra seconds to the call.

DTMF playout for IPDC

TAOS supports Dual Tone Multi-Frequency (DTMF) digit playout signalled by the IPDC STN message. It plays the DTMF digits utilizing a Digital Signal Processor (DSP) on the line card or a MultiDSP card. A DSP on the MultiDSP card can be associated with a VoIP call route, so Softswitch can direct a MultiVoice Gateway to play the DTMF digits during an active VoIP call.

Note Refer to *Level 3 Communications, Internet Protocol Device Control (IPDC), Revision 0.15* specification for an explanation of all messages and tags that are referred to in the following sections.

MultiVoice Gateway Configuration
DTMF playout for IPDC

IPDC messages that support DTMF playout

Following are IPDC messages (as defined in IPDC specification *Level 3 Communications, Internet Protocol Device Control (IPDC), Revision 0.15)* that support DTMF playout.

Table 2-20. IPDC messages supporting DTMF playout

IPDC message	Tag	Tag values
STN	0x49 (Tone Type)	Specify 0x01 (DTMF tone) to play DTMF digits.
	0x4A (Apply/Cancel Tone)	The following value is supported: 0x00 (Apply tone)
	0x32 (Num Tones)	The value associated with this tag indicates the number of DTMF digits to be played out.
	0x33 (Tone String)	The value associated with this tag contains the DTMF digits to be played out.
ASTN	0x36 (Completion Status)	The following values are returned: • 0x00 (Operation successful) • 0x01 (Operation failed) • 0x03 (Operation started)

Basic call flow for DTMF digits played during a packet call

The following call flow illustrates how the STN message is used to play DTMF digits during an active packet call. It shows a Softswitch setting up a packet call and then arbitrarily requesting that the MultiVoice Gateway at the far end play three DTMF digits: 1, 2 and 3.
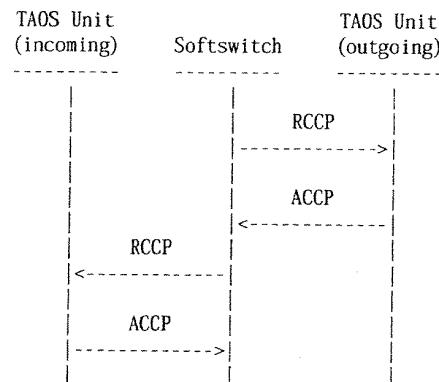
```
    TAOS Unit                    TAOS Unit
    (incoming)    Softswitch     (outgoing)
    ----------    ----------     ----------
        |             |              |
        |             |     RCCP     |
        |             |------------->|
        |             |              |
        |             |     ACCP     |
        |             |<-------------|
        |     RCCP    |              |
        |<------------|              |
        |             |              |
        |     ACCP    |              |
        |------------>|              |
        |             |              |
```

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MultiVoice Gateway Configuration
DTMF playout for IPDC

```
                     |        RTP       |                  |
                     |<- - - - - - - - - - - - - - - - ->|voice exchange
                     |                  |                  |begins
                     |                  |       STN        |
                     |                  |- - - - - - - - ->|Tag 0x33 = "123"
                     |                  |      ASTN        |
                     |                  |<- - - - - - - - -|operation started
                     |                  |                  |Tag 0x36 = 0x03
                     |                  |                  |
                     |                  |                  |"1" played
                     |                  |                  |
                     |                  |                  |"2" played
                     |                  |                  |
                     |                  |                  |"3" played
                     |                  |      ASTN        |
                     |                  |                  |operation successful
                     |                  |<- - - - - - - - -|Tag 0x36 = 0x00
                     |                  |                  |
```
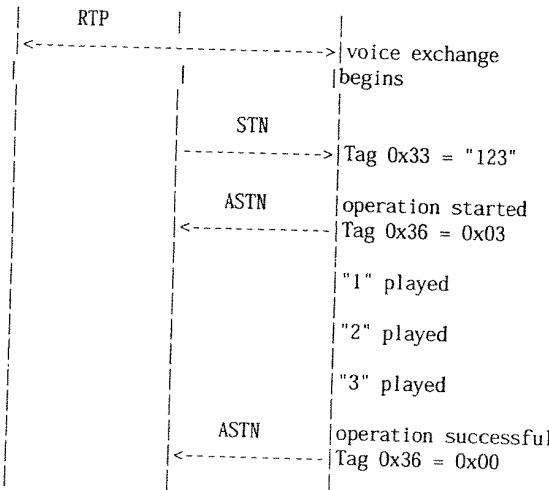
Call flow for out-of-band DTMF transport

The following call flow illustrates how DTMF playout is used in conjunction with in-call DTMF detection to achieve true out-of-band DTMF transport using IPDC signalling.

In this example, the call originator enters two DTMF digits, 1 and 2, after the voice exchange has begun. When performing out-of-band DTMF transport, it is necessary to remove the entered DTMF from the RTP stream. To do so, set the voip profile on the MultiVoice Gateway as follows:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set dtmf-tone-passing = dtmf-tone-passed-outofband
admin> write
VOIP/{ 0 0 } written
```

The call flow for out-of-band DTMF transport is as follows:

```
           TAOS Unit                      TAOS Unit
           (incoming)      Softswitch     (outgoing)
          - - - - - - -   - - - - - - -   - - - - - - -
                |               |               |
                |               |               |
                |               |               |
                |               |      RCCP      |
                |               |- - - - - - - ->|
                |               |               |
                |               |      ACCP      |
                |               |<- - - - - - - -|
DTMF detection  |      RCCP      |               |
enabled         |<- - - - - - - -|               |Tag 0x75 = 0x01
                |               |               |
                |      ACCP      |               |
                |- - - - - - - ->|               |
                |               |               |
```

MultiVoice Gateway Configuration
DTMF playout for IPDC

```
                          |         |         |
                          |   RTP   |         |
                          |<- - - - - - - - - - - - - - - - - ->|voice exchange
                          |         |         |begins
                          |         |         |
        user enters       |   NTN   |         |
        DTMF digit "1"    |- - - - - - - - - ->|         |Tag 0x33 = "1"
                          |         |         |
                          |         |   STN   |
                          |         |- - - - - - - - - ->|Tag 0x33 = "1"
                          |         |         |
                          |         |   ASTN  |
                          |         |<- - - - - - - - - -|Tag 0x36 = 0x03
                          |         |         |
                          |         |         |
                          |         |         |"1" played
                          |         |         |
                          |         |   ASTN  |
                          |         |<- - - - - - - - - -|Tag 0x36 = 0x00
        user enters       |   NTN   |         |
        DTMF digit "2"    |- - - - - - - - - ->|         |Tag 0x33 = "2"
                          |         |         |
                          |         |   STN   |
                          |         |- - - - - - - - - ->|Tag 0x33 = "2"
                          |         |         |
                          |         |   ASTN  |
                          |         |<- - - - - - - - - -|Tag 0x36 = 0x03
                          |         |         |
                          |         |         |
                          |         |         |"2" played
                          |         |         |
                          |         |   ASTN  |
                          |         |<- - - - - - - - - -|Tag 0x36 = 0x00
                          |         |         |
```
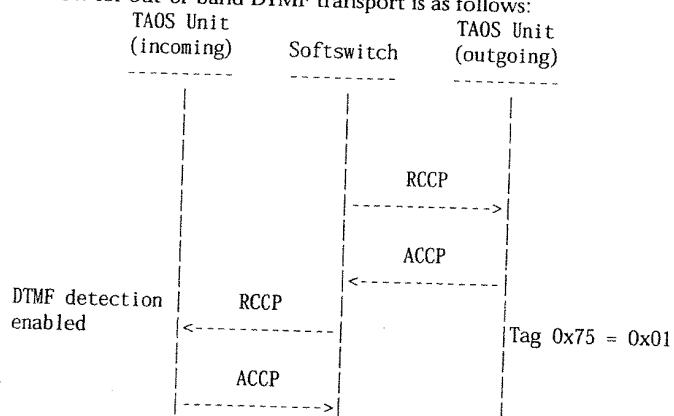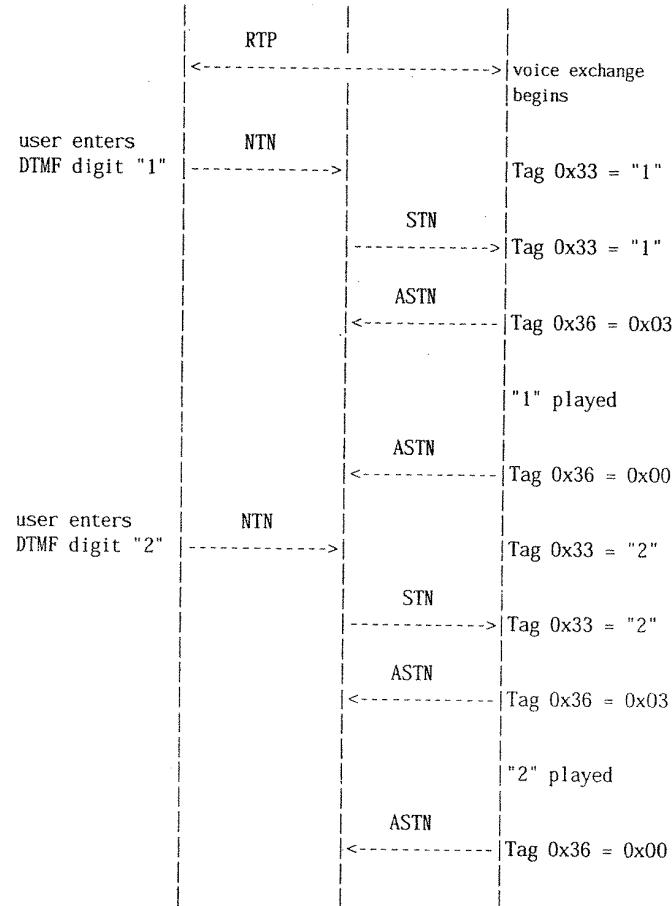
An STN request for DTMF playout over VoIP is accepted only for an active VoIP call in voice exchange mode. It is rejected for an active VoIP call that is performing pre-call DTMF collection, playing a pre-call announcement or playing a break-in announcement.

Only the Apply command is allowed—Cancel is not allowed.

The operation is allowed regardless of the setting of the dtmf-tone-passing parameter in the voip profile (that is, inband, out-of-band or rfc2833).

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Error Handling

In addition to the existing conditions whereby an STN request can be rejected, the following error responses are generated:

Table 2-21. Error handling

Softswitch will receive:	For any of the following:
MRJ with Tag 0xFE (Cause Code) = 0x65 (Wrong Message For State)	• STN while a VoIP call is performing pre-call DTMF detection. • STN while a VoIP call is playing a pre-call voice announcement. • STN while a VoIP call is playing a break-in voice announcement.
ASTN with Tag 0x36 (Completion Status) = 0x01 (Operation Failed)	• STN with Tag 0x4A (Apply/Cancel Tone) = 1 (Cancel)

IPDC country-specific call-progress tone playout for VoIP

IPDC defines a specification for playing tones, such as DTMF or arbitrary frequencies towards the PSTN. TAOS provides a means for signaling one of several call progress tones (for example, dial tone, alerting, busy, network busy and unobtainable) using a MultiDSP slot card.

You can also configure the unit to generate a country-specific call progress tone. Country-specific call progress tones can be played as

- Local (to the calling party) playout of remote country-specific call progress tones, which is explained in this section.

- Carry back the call progress tones via RTP, which is already supported in the current IPDC VoIP implementation in TAOS using existing IPDC messaging.

STN message that supports country-specific call-progress tones

A tone type that has been added to the STN (Send tones or announcement) message includes a a country identifier.tone string that identifies one of the call progress tones. TAOS uses the tone string and country identifier to generate a country-specific call progress tone.

The STN message that supports generating country-specific call-progress tones is defined in IPDC specification *Level 3 Communications, Internet Protocol Device Control (IPDC), Revision 0.15.*

Tag 0x49 (tone type)

The proprietary tag value of 0x41 (Call Progress Tone) has been added to the tone type tag 0x49. The tag value minimizes conflict with any future tag values that might be added in the future by the IPDC community at large. Value 0x41 signals that the STN message contains a request to apply or cancel a call progress tone.

MultiVoice Gateway Configuration
IPDC country-specific call-progress tone playout for VoIP

Associated with this value for tag 0x49 is a set of additional values used to indicate the type of call progress tone to play. These values are specified in tag 0x33 (tone string) whenever 0x49 has value 0x41, as follows:

Call progress tone	Tag 0x33 value
Dial tone	"1"
Alerting tone	"2"
Busy tone	"3"
Network busy tone	"4"
Unobtainable tone	"14"

The next available non-proprietary tag value for tag 0x49 is 0x08.

Note TAOS also provides support for two other call progress tones (that is, pin tone and error tone). However, these tones are not country-specific and are relevant only for the H.323 VoIP implementation in TAOS.

Tag 0xC1 (Country Identifier)

The proprietary tag, 0xC1, has also been added to the STN message whenever the STN message contains a request to apply or cancel a call progress tone (that is, when tag 0x49 has the value 0x41).

The following table lists the value of the 0xC1 tag generated in the STN message according to the setting of the country parameter in the system profile:

Parameter setting	Tag 0xC1 value
argentina	0x01
australia	0x02
belgium	0x03
china	0x04
costa rica	0x05
finland	0x06
france	0x07
germany	0x08
hong kong	0x09
italy	0x0A
japan	0x0B
korea	0x0C
mexico	0x0D
netherlands	0x0E
new zealand	0x0F
singapore	0x10
spain	0x11
sweden	0x12

Parameter setting	Tag 0xC1 value
switzerland	0x13
uk	0x14
us	0x15
brazil	0x16

The value of tag 0xC1 is used as the country of origin when determining the frequency, duration, and cadence of the call progress tone. If tag 0xC1 is omitted, the value of the country parameter in the system profile is used as the country of origin.

STN/ASTN usage

The semantics of the STN and ASTN (Completion result of STN command) messages remain the same as for the other tones and announcements.

- To start a call progress tone, send an STN message with tag 0x4A set to 0x00 (Apply Tone).

- To stop a call progress tone, send an STN message with tag 0x4A set to 0x01 (Cancel Tone).

All of the above five call progress tones play continuously until explicitly canceled.

Note If the universal gateway is unable to accept an STN call progress tone request, it responds with an ASTN with status 0x01 (operation failed).

Sample Call Flows — ringing, then answered scenario

If you network consists of two universal gateways, one in the United Kingdom (MVGW-UK) and one in France (MVGW-FRANCE), with one Softswitch, and the calling party is in the UK and the called party is in France, to allow the calling party to hear an alerting tone specific to France, Softswitch sends MVGW-UK an STN with tag 0xC1 = 0x07, tag 0x49 = 0x41, tag 0x33 = "2", tag 0x4A = 0x00 and other tags as required for STN.

Softswitch received tag 0xC1 = 0x07 from MVGW-FRANCE when MVGW-FRANCE sent an NSUP to the Softswitch. .

Note The NSUP message is unsupported.

MultiVoice Gateway Configuration
IPDC country-specific call-progress tone playout for VoIP
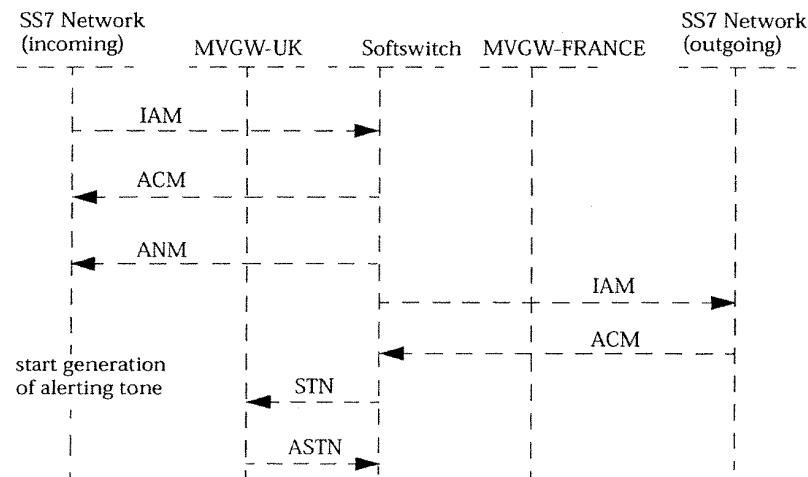
Message contents:

Softswitch->MVGW: (STN - apply "alerting" call progress tone)

```
        Protocol=0x4b, Correlator (4): 00000000
        Message: 0x0073
        Tag ID=0x07, Data (2): 00 01
        Tag ID=0x0d, Data (2): 00 01
        Tag ID=0x15, Data (2): 00 01
        Tag ID=0x49, Data (1): 41
        Tag ID=0x4a, Data (1): 00
        Tag ID=0x32, Data (1): 01
        Tag ID=0x33, Data (1): 32
        Tag ID=0xc1, Data (1): 07
```

MVGW->Softswitch: (ASTN - command started)

```
        Protocol=0x4b, Correlator (4): 00000000
        Message: 0x0074
        Tag ID=0x07, Data (2): 00 01
        Tag ID=0x0d, Data (2): 00 01
        Tag ID=0x15, Data (2): 00 01
        Tag ID=0x36, Data (1): 03
```

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When the called party answers, Softswitch stops the alerting tone by sending MVGW-UK an STN with the same tags as above except tag 0x4A = 0x01. The packet call can then be set up.
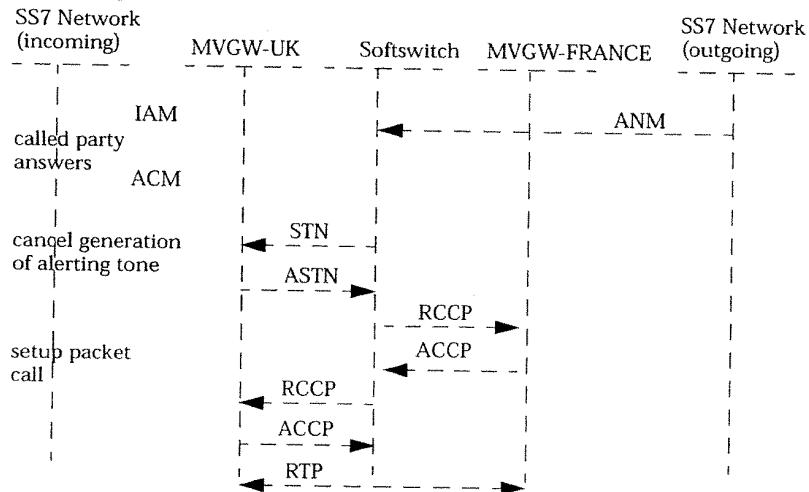
Message contents:

Softswitch->MVGW: (STN - cancel "alerting" call progress tone)

```
Protocol=0x4b,  Correlator (4): 00000000
Message: 0x0073
Tag ID=0x07, Data (2): 00 01
Tag ID=0x0d, Data (2): 00 01
Tag ID=0x15, Data (2): 00 01
Tag ID=0x49, Data (1): 41
Tag ID=0x4a, Data (1): 01
Tag ID=0x32, Data (1): 01
Tag ID=0x33, Data (1): 32
Tag ID=0xc1, Data (1): 07
```

MVGW->Softswitch: (ASTN - command completed)

```
Protocol=0x4b,  Correlator (4): 00000000
Message: 0x0074
Tag ID=0x07, Data (2): 00 01
Tag ID=0x0d, Data (2): 00 01
Tag ID=0x15, Data (2): 00 01
Tag ID=0x36, Data (1): 00
```

Sample Call Flows — busy, then hangup scenario

If call setup never progresses to the point where a packet call is actually setup (that is, an RCCP is sent and accepted by the universal gateway), but at least one call progress tone has been generated, and it is time to take down the call, then when VoIP call persistence is enabled on the gateway, it is necessary for the Softswitch to send the

MultiVoice Gateway Configuration
IPDC country-specific call-progress tone playout for VoIP

gateway an RCR at the end of the call so that the gateway may free the resources that were used to generate the call progress tone or tones.

To see if Voip call persistence is enabled, check the setting of the ss7voip-call-persistence in the voip profile:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> list
[in VOIP/{ 0 0 }]
...
ss7voip-call-persistence = yes
...
```

In the following call flow, the called party is busy. The Softswitch directs the universal gateway to generate a busy tone specific to France. After a short while, the calling party hangs up. Since VoIP call persistence is enabled, the Softswitch must send the gateway an RCR to allow it to free the resources associated with call progress tone generation when VOIP call persistence is enabled.
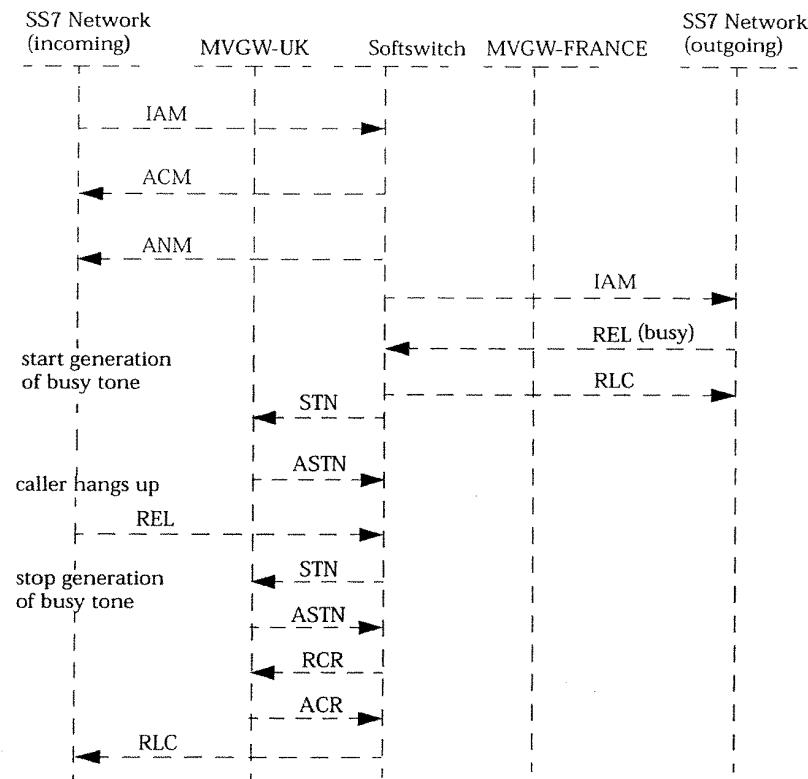
The RCR allows the universal gateway to free the resources (for example, MultiDSP slot card) associated with the VoIP call route that was setup for the call progress tone generation when the first STN was received.

Note If VoIP call persistence was disabled, the RCR would not be needed.

VoIP Call Configuration

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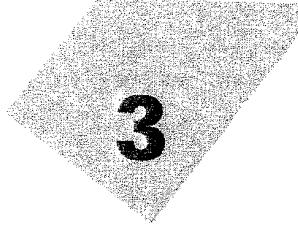
The voip profile

The voip profile configures call-performance features and manages H.323 call processing. The following VoIP call-performance features are configured through the voip profile:

- The type of voice compression and coding to use for VoIP calls.
- Enabling use of a fixed or dynamic jitter buffer.
- Enabling silence detection and comfort noise generation.
- Adjust the relative lever of silence suppression.
- Modifying the Type-of-Service (ToS) byte for UDP packet processing.
- Modify the maximum number of calls a TAOS unit processes.

The following H.323-specific functions are configured through the voip profile:

- The IP address of the H.323 gatekeeper —MultiVoice Access Manager (MVAM).
- The IP address for a secondary H.323 gatekeeper (MVAM).
- Adjust the frequency and time intervals when a TAOS unit must register with MVAM.
- Enable Personal Identification Number (PIN) collection for authentication by MVAM.
- Enable single-stage dialing.
- Enable progress tone cut-through from the distant PSTN on the local TAOS unit.
- Adjust the amount of time a caller has to dial a telephone number.
- Support for multiple logical gateways.
- Configure voice announcements in place of call-progress tones.
- Enable rerouting of blocked calls back out over the PSTN.

VoIP Call Configuration
The voip profile

- Enable out-of-band pass through of PSTN call-progress tones.
- Block Caller ID.
- Configure the call interdigit timer.
- Delay alerting the PSTN about active calls until the connection is completed across the MultiVoice network.
- Enable transparent modem signaling.

The MultiVoice call configuration options are located in the voip profile:

Parameter	Setting
voip-index*	Identifies the voip profile by telephone number. This subprofile uses the telephone number (DNIS) associated with an inbound trunk and the called destination to control routing and processing of VoIP calls. The default voip profile, voip { 0 0 }, is a system wide profile used for processing all VoIP calls. Each DNIS-specific VoIP profile can contain settings which apply only to calls received on the associated trunk.
gk-mlg-control	Provides support for partitioning a single MultiVoice Gateway into multiple logical gateways. Partitioning call control lets the H.323 Gatekeeper perform call-specific administration on a call-by-call basis.
gatekeeper-ip	Identifies the H.323 gatekeeper associated with this TAOS unit. This gatekeeper, usually running MVAM, performs H.323 registration, admission and status reporting for this MultiVoice Gateway. **Note** After changing the default value of 0.0.0.0 to an IP address, you need to reset the TAOS unit.
vpn-mode	Enables or disables H.323 call authentication on a TAOS unit. When authentication is enabled, a TAOS unit prompts for a user- entered Personal Identification Number (PIN).
packet-audio-mode	Selects the default audio coder/decoder (codec) used to process analog voice, received from the public switched telephone network (PSTN) and packetized voice, for transmission across the packet network.
frames-per-packet	Sets the number of voice frames transmitted in a single RTP packet, across the IP network, between two MultiVoice Gateways.
tos-options subprofile	Sets the requested Type-of-Service (ToS) processing priority for RTP packets sent across the IP network between two MultiVoice Gateways.

Parameter	Setting
silence-det-cng	Setting this parameter to yes enables silence suppression. When enabled, the sending side of the call uses silence suppression during background noise conditions. On the receiving side, suppressed sections will always be filled in with locally generated noise. During those silent periods, the local TAOS unit will generate background (comfort) noise to assure the caller that the call is still connected.
gatekeeper-ip-sec	Identifies a secondary H.323 gatekeeper associated with this TAOS unit. This gatekeeper, usually running MVAM, performs H.323 registration, admission and status reporting for this TAOS unit, if the TAOS unit can't register with the gatekeeper specified by the gatekeeper-ip parameter.
gatekeeper-keepalive	Sets the time interval between attempts a TAOS unit makes to reregister with a system running MVAM, following the initial registration. This value equals the wait time, in seconds, between each attempt to reregister.
registration-retries	Sets the number of attempts a TAOS unit makes each time it executes keepalive registration. Since a MultiVoice Gateway may not successfully register on its first attempt, the value for this parameter represents the number of repeated registration attempts a gateway makes during a registration cycle.
registration-retry-timer	Sets the time interval between each registration attempt a TAOS unit makes with MVAM. This sets the pause, in seconds, between each registration attempt specified by the registration-retries parameter.
primary-retries	Sets the number of attempts a TAOS unit makes whenever it tries to reregister with the MultiVoice Access Manager at gatekeeper-ip after registering with the gatekeeper at gatekeeper-ip-sec. Since a MultiVoice Gateway may not successfully register on its first attempt, the value for this parameter represents the number of repeated registration attempts a gateway makes during a reregistration cycle.
ena-adap-jitter-buffer	Changes the jitter buffer mode to either adaptive or fixed for VoIP calls. When the adaptive mode is selected, the jitter buffer will range in size between the values set for max-jitter-buffer-size and one packet.
max-jitter-buffer-size	Sets the maximum jitter buffer size for VoIP calls when the TAOS unit is configured to perform adaptive call jitter buffering. When using adaptive mode, the jitter buffer may increase to accommodate the entered number of audio packets, based on the incoming packet arrival statistics (jitter).

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VoIP Call Configuration
The voip profile

Parameter	Setting
initial-jitter-buffer-size	Sets the initial jitter buffer size for VoIP calls when the TAOS unit is configured to perform adaptive call jitter buffering. When using either adaptive or fixed mode, the jitter buffer is set to initial-jitter-buffer-size at start-up.
maxcalls	Sets the maximum number of VoIP calls a TAOS unit can process simultaneously, by limiting the number of Digital Signal Processors (DSPs) available.
cut-thru-enable-nearend	Enables or disables transmission of call-progress tones from the far-end public switched telephone network (PSTN) across the IP network to the local TAOS unit, for play out to the caller.
single-dial-enable	Enables or disables single-stage dialing for VoIP calls when MultiVoice is used to perform H.323 call processing.
h323-voice-ann-enabled	Enables or disables play out of voice announcements to report call-progress for VoIP call processing.
voice-ann-dir	Identifies the directory on the external flash memory where voice announcement files are stored for call-progress reporting.
call-inter-digit-timeout	Sets the limit on how long the TAOS unit waits for a caller to enter a single digit when using two-stage dialing, and when entering digits during a call.
silence-threshold	Sets the relative threshold for silence suppression to compensate for background noise levels when silence suppression is enabled (silence-det-cng=yes).
dtmf-tone-passing	Specifies how DTMF tones detected at the ingress gateway are transmitted to the egress gateway. Specify one of the following values: • inband– The near-end gateway passes PSTN-generated DTMF digits and tones as part of the voice processing stream. These tones are compressed by the selected audio codec and transported across the IP network using UDP packets. • outofband–The near-end gateway passes PSTN-generated DTMF digits and tones across the network using non-UDP packets. Once received at the far end, the digits are played out to the local PSTN/caller. • rtp–DTMF tones are transferred and passed via another channel to the decoding DSP, according to the RFC2833 standard. **Note** Both near-end and far-end gateways must have the same setting.

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Parameter	Setting
rt-fax-options subprofile	Enables or disables real-time fax call processing.
call-hairpin	Enables or disables attempts to re-route blocked VoIP calls from a TAOS unit using a connection to the local public switched telephone network (PSTN).
call-keep-alive-timeout	Sets the time interval that a MultiVoice Gateway will wait before polling a remote gateway and/or client during a VoIP call, to verify that they are still functioning, and reachable over the IP network.
clid-suppress	Enables or disables blocking transmission of the Calling Line ID (CLID) associated with a call to the local PSTN by the MultiVoice Gateway. The gateway can send a blocked number message or substitute CLID received from MVAM to the PSTN.
true-connect-enable	Enables or disables a TAOS unit delay reporting a call connected to the local PSTN until both parties in a call are connected.
g711-transparent-data	Enables or disables detection of high-speed fax/modem signals on a VoIP channel, and enables fax/modem transmission in a transparent mode using the G.711 codec at 64Kbps.
allow-g711-fallback	Enables or disables fallback to the G.711 audio codec when either H.323 end point (such as, a gateway or terminal) involved in a VoIP call does not support the audio codec designated by the packet-audio-mode parameter.
allow-coder-fallback	Enables or disables fallback to a negotiated codec when either H.323 end point (such as, a gateway or terminal) involved in a VoIP call does not support the audio codec designated by the packet-audio-mode parameter.
trunk-quiesce-enable	Enables or disables automatic trunk deactivation when the MultiVoice Gateway is unavailable to process calls.
early-ringback-enable	Enables or disables local generation of a ringback tone by a TAOS unit as soon as the call-setup begins on the far-end MultiVoice Gateway.
trunk-prefix-enable	Enables or disables assignment of the egress trunk group by the ingress TAOS unit. The MultiVoice Gateway prepends the trunk group number associated with the entry (ingress) T1/E1 trunk to the destination telephone number sent to the exit (egress) gateway or call signaling entity. The egress gateway connects the call to the PSTN using a DS0 assigned to the designated trunk group.

VoIP Call Configuration
The voip profile

Parameter	Setting
operator-assist	Allows callers to request operator assistance during the dialing phase of a MultiVoice call. A TAOS unit can be assigned a dial string, up to five digits long, that may be entered by a caller in order to connect that caller to an operator.
sequential-call-enable	If a caller must enter a PIN to authenticate MultiVoice calls, to dial subsequent VoIP calls without reentering the PIN, as long as the connection between the PSTN and near-end MultiVoice Gateway is not terminated.
next-call	A new call can be initiated by dialing a string (for example, **9) as specified in the next-call parameter in the voip profile. Once the dialing string has been entered, the user hears a dial tone and can then proceed to enter the entire 7- or 10-digits (if the call is a long-distance call) number.
ss7voip-call-persistence	Setting this to yes causes VoIP call route to persist across VoIP-related IPDC requests for a given call (e.g., LTN, STN, RCCP and RMCP) until the call is released (via RCR). Setting this to no disables the feature and the VoIP call route exists only for the life of the single IPDC request, or in the case where an announcement (STN) and DTMF detection (LTN) are overlapping, after the announcement or the DTMF detection has completed, whichever occurs last. Enabling VoIP call persistence results in faster call setup and call processing times for SS7 VoIP calls initiated through IPDC.
faststart-enable	Setting this to yes results in much faster call setup in the network than that provided by the standard H.245 procedure. In situations in which fast connect is unsuccessful, the call is automatically set up using standard H.245 procedures instead.
rtpqos-polling-enable	Setting this to yes generates RTP QoS statistics periodically, through a polling parameter. RTP QoS periodic statistics (such as end-of-call statistics) are sent to the IPDC protocol (this function is not dependent upon the enabling of either RTP QoS polling or Call Logging).
signaling-tos subprofile	Enables configuring DSCP values for marking H.323 signaling packets.

Parameter	Setting
pstn-attribute subprofile	Changes the way an egress MultiVoice Gateway manages call signaling with the switched network. For example:

- Delivery of Q.931/Q.850 cause codes are transparent when received from the PSTN by the far-end MultiVoice Gateway to the near-end MultiVoice Gateway.

- Bearer capabilities sent in the Q.931 Setup message by the far-end MultiVoice Gateway for outbound calls to the switched network are configurable.

- Reporting of Q.931 Progress Indicator information element (IE) in the Proceeding and Alerting message by the near-end MultiVoice Gateway to the switched network is configurable.

If you are configuring a TAOS unit to work in an H.323 environment, you must provide an IP address for the gatekeeper-ip parameter to process VoIP calls. The IP address points to the computer running MVAM that performs all of the H.323 gatekeeper functions for the TAOS unit. The TAOS unit can process VoIP calls over most IP networks using the factory defaults for the remaining voip profile parameters.

Note In this release, you may not change the values contained in DNIS-specific voip profiles. The TAOS unit globally applies the values set in the default voip profile to all VoIP calls. DNIS-specific voip profiles are only used to simplify internal processing and routing of VoIP calls.

Creating DNIS-specific voip profiles

User-defined voip profiles are used to map incoming calls by identifying all calls associated with a specific Dialed Number Identification Service (DNIS) string as VoIP calls. See "Using DNIS-specific trunk mappings" on page 2-15.

For example, if a user created the following voip profiles:
```
admin> dir voip
    46   12/23/1998 09:48:55   { 0 0 }
    31   12/18/1998 09:50:06   { 8093190 0 }
    31   12/18/1998 10:07:16   { 8903190 0 }
```

The TAOS unit processes all calls from the PSTN with these DNIS strings as VoIP calls. The voip-index subprofile distinguishes between the default voip profile, voip {0 0}, and any user-created voip profiles:
```
admin> list voip-index
[in VOIP/{ 8903190 0 }:voip-index
gateway-access-number = 8903190
far-end-number = 0
```

VoIP Call Configuration
Configuring call-performance parameters

This subprofile includes the following parameters:.

Parameter	Specifies
gateway-access-number	This is the Dialed Number Identification String (DNIS) passed from the PSTN associated with the in-bound telephone number used to access the TAOS unit. If the TAOS unit is configured to perform two-stage dialing of VoIP calls, this would be the telephone number dialed to access the TAOS unit from the PSTN.
far-end-number	This value should always be set to 0.

To set the values for these parameters, use the new and write commands to create user-defined voip profiles:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> new voip { 8903190 0 }
VOIP/{ 8903190 0 } read
admin> write
VOIP/{ 8903190 0 } written
admin> dir voip
    46 12/23/1998 09:48:55 { 0 0 }
    31 12/18/1998 10:07:16 { 8903190 0 }
```

Note You may create DNIS-specific voip profiles by changing the value of the gateway-access-number parameter using the set command; but only if no other changes have been written to the voip { 0 0 } profile.

Configuring call-performance parameters

The call-performance parameters control how a MultiVoice Gateway processes calls received from the PSTN. This group of parameters affects allocation of packet network bandwidth for each call, the allocation of DSP assets for each call, and subsequently, the number of VoIP calls that a TAOS unit can process simultaneously. The following voip profile parameters handle VoIP call-performance functions:

- packet-audio-mode
- frames-per-packet
- allow-g711-fallback
- allow-coder-fallback
- silence-det-cng
- silence-threshold
- ena-adap-jitter-buffer
- max-jitter-buffer-size
- initial-jitter-buffer-size
- tos-options
- maxcalls
- faststart-enable

Configuring voice compression

Voice is transmitted across an IP network as compressed audio frames, which are compressed/decompressed by the TAOS unit. The packet-audio-mode parameter specifies which default audio codec (coder/decoder) packs (and unpacks) analog speech into digital audio frames. You may enter any of the following values representing these supported audio codecs:

Parameter value	Specifies
g711-ulaw	G.711 µ–law
g711-alaw	G.711 a-law
g729	G.729(A)
g723	G.723.1 running at 5.3kps
g723-6.4kps	G.723.1 running at 6.4kps
g728	G.728
frgsm	Full-rate GSM

The default value for the packet-audio-mode parameter is g711-ulaw. The following example illustrates how to set the audio codec used for processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set packet-audio-mode = g729
admin> write
VOIP/{ 0 0 } written
```

Changes to the setting take effect with the next call. The packet-audio-mode parameter has the following dependencies:

- TAOS units configured with 96-port MultiDSP slot cards (APX8-SL-96DSP) support using only the G.711 or G.729(A) audio codecs.

- This parameter does not prevent other supported audio codecs from being dynamically selected during call-setup.

- The silence-det-cng parameter is ignored when using G.711 a-law or G.711 µ–law. For details, see "Configuring silence detection and comfort noise generation" on page 3-13.

- When either G.723 or G.723-6.4kps codec is specified:

    - silence-det-cng may be enabled or disabled for 6.4Kbps processing only (packet-audio-mode=g723-6.4kps).

    - Comfort noise generation may be enabled or disabled for 5.3Kbps processing. With comfort noise enabled, the 5.3Kbps can decode silence detection and suppression packets. Silence detection/suppression cannot be selected for 5.3Kbps processing since it will not encode silence. This is in accordance with standards.

    - Comfort noise generation cannot be enabled for 5.3Kbps processing unless the adaptive jitter buffer is disabled.

    - Silence detection/suppression cannot be enabled for 6.4Kbps processing unless the adaptive jitter buffer is disabled.

VoIP Call Configuration
Configuring call-performance parameters

– Adaptive jitter buffer processing can be enabled when silence detection/suppression is disabled.

– The actual maximum size of the adaptive jitter buffer is limited to nine frames per packet for G.723.1 both rates.

G.728 codec support

G.728 is an audio codec based on Low-Delay Code Excited Linear Prediction (LD-CELP). G.728 provides toll-quality audio at a bit rate of 16Kbps. With a frame size of only 2.5 milliseconds, G.728 also has a very low delay. Although the MultiVoice implementation of G.728 uses a frame size of 5 milliseconds, the bitstream from the audio codec is the same as described in the ITU-T standard and can thus be decoded by any G.728 decoder.

When the G.728 codec is selected, the MultiVoice Gateway attempts to determine if the G.728 codec is supported by the other gateway during H.245 capability negotiation. If both sides agree to use G.728 as the preferred codec, both gateways use G.728 to compress and decompress audio after the H.245 open logical channel message is exchanged.

Although MultiVoice uses a 5-millisecond frame for G.728 processing, it is compatible with any third-party G.728 decoder. However, if a MultiVoice Gateway attempts to communicate with a third-party VoIP gateway transmitting an odd number of 2.5 millisecond frames per IP packet, the call fails.

When you enable G.728 audio processing (packet-audio-mode=g728), the silence-det-cng parameter in the voip profile must be set to no (its default value). The following commands enable G.728 processing:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read

admin> set packet-audio-mode = g728

admin> set silence-det-cng = no

admin> write
VOIP/{ 0 0 } written
```

G.723.1 codec support

When G.723.1 codec is selected

• silence-det-cng can be enabled or disabled for 6.4Kbps processing only by setting the packet-audio-mode parameters as follows:

packet-audio-mode=g723-6.4kps

• Comfort noise generation can be enabled or disabled for 5.3Kbps processing by setting the packet-audio-mode to g723-5.3kps. When comfort noise is enabled, the 5.3Kbps setting allows silence detection and suppression packet decoding. Silence detection/suppression cannot be configured directly for 5.3Kbps processing since it will not encode silence. This is in accordance with standards.

• Comfort noise generation cannot be enabled for the packet-audio-mode 5.3Kbps setting processing unless adaptive jitter buffering is disabled and the ena-adap-jitter-buffer is set to no.

• Silence detection/suppression cannot be enabled for 6.4Kbps processing unless the adaptive jitter buffer is disabled.

- Adaptive jitter buffer processing can be enabled for:
    - 6.4Kbps processing when silence detection/suppression is disabled,
    - 5.3Kbps processing when comfort noise generation is disabled.
- The actual maximum size of the adaptive jitter buffer is limited to nine frames per packet for G.723.1 both rates.

Full-Rate GSM codec support

Full-Rate GSM (Global System for Mobile Communications) is a voice encoder/decoder standard for cellular communications. Full-Rate GSM compresses the speech samples from 64Kbps PCM to 13.2Kbps, requiring less network than G.711 a-law or G.711 μ–law. It is the standard followed by European, Japanese, and Australian cellular communications systems, and is supported by certain Web phone applications.

Full Rate GSM uses a speech frame size of 160 samples (20msec) and the encoder produces 33 bytes per frame. The decoder produces 160 samples (20msec) of speech from the 33-byte encoder output.

The Full Rate GSM audio codec is defined by ETSI Recommendation GSM 06.10, *GSM Full Rate Speech Transcoding,* (Feb. 1992), European Telecommunications Standards Institute. Full Rate GSM also supports Silence Detection and Comfort Noise Generation, as defined by the ETSI Recommendation GSM 06.12, *Comfort Noise Generation,* (Feb. 1992), European Telecommunications Standards Institute and ETSI Recommendation GSM 06.12, *Discontinuous Transmission* (Feb. 1992), European Telecommunications Standards Institute.

A MultiVoice Gateway reports Full-Rate GSM during H.245 capability negotiation. If both H.323 end points (such as, a MultiVoice Gateway and a PC, or two MultiVoice Gateways) choose Full-Rate GSM as the preferred codec, then, after opening the H.245 logical channel between both H.323 end points, Full-Rate GSM is used for processing the VoIP call. Full-Rate GSM is encoded as a standard audio capability.

Configuring voice packet size

The number of compressed audio frames assigned to each RTP packet used to transport voice across the IP network is controlled by the frames-per-packet parameter. You can assign a value ranging from 1 to 10 packets. The default is 4.

The following example illustrates how to change the number of audio frames assigned to each RTP packet for processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set frames-per-packet = 6
admin> write
VOIP/{ 0 0 } written
```

Assigning a lower value to the frames-per-packet parameter reduces the delay and distortion introduced into any given voice call. But using a lower value may also degrade performance as the number of RTP packets processed for a voice call increases.

Note When a different audio codec is dynamically selected during call-setup, a TAOS unit uses the default value of four frames per RTP packet to process that call.

For more information on MultiVoice packet processing see Appendix A, "MultiVoice Packet Processing."

Configuring audio codec negotiation

Voice is transmitted across an IP network as compressed audio frames. The packet-audio-mode parameter in the default voip profile specifies the preferred audio codec used by the gateways to compress and uncompress analog speech and digital audio frames.

You can use the following parameters (shown with default values) to specify how the system behaves when the preferred codec is not supported for all VoIP, fax, and transparent modem calls:

```
[in VOIP/{ 0 0 }]
allow-g711-fallback = yes
allow-coder-fallback = yes
```

Parameter	Setting
allow-coder-fallback	Overrides fallback to a negotiated codec when either of H.323 end points (such as a gateway or terminal) involved in a VoIP call do not support the audio codec designated by the packet-audio-mode parameter.
	When allow-coder-fallback=no, the ingress gateway rejects the call if it is unable to connect the call using the preferred codec. If this parameter is set to no, the allow-g711-fallback parameter has no effect.
	When allow-coder-fallback=yes, the ingress gateway negotiates an alternate audio codec with the destination gateway during call capabilities setup. Yes is the default setting.
allow-g711-fallback	Overrides fallback to the G.711 audio codec when either of H.323 end points (such as a gateway or terminal) involved in a VoIP call does not support the audio codec designated by the packet-audio-mode parameter.
	If allow-coder-fallback=yes, setting allow-G711-fallback=no prevents the ingress gateway from selecting the G.711 codec when negotiating call capabilities. In this case, the system terminates the call if G.711 is the only available choice and it is not the preferred codec.
	When allow-G711-fallback=yes, the ingress gateway may negotiate using the G.711 audio codec with the destination gateway during call capabilities setup. yes is the default setting.

Normally, an H.323 stack advertises a list of supported audio codecs. If the preferred codec is present on a list received from a far-end gateway, that codec is always selected. Otherwise, the system selects an alternate codec that matches one from its supported list.

Modifications made to these parameters become effective with the next VoIP call.

The following example illustrates how to allow fallback to any supported audio codec, except G.711, when processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set allow-coder-fallback = yes
admin> set allow-g711-fallback = no
admin> write
VOIP/{ 0 0 } written
```

Configuring silence detection and comfort noise generation

A TAOS unit can be configured to detect periods of silence during voice calls, suppress transmission of voice packets containing silence, and generate white (comfort) noise to assure the user that a call is still connected during silent periods.

The silence-det-cng parameter enables or disables the feature. You can prevent silence frames from being passed across the packet network, reducing the effective bandwidth of the VoIP call. During those silent periods, the local TAOS unit generates background (comfort) noise to assure the caller that the call is still connected during these silent periods. You can assign the following values to the silence-det-cng parameter:

Parameter value	Specifies
yes	Silence frames are not passed across the IP network by the TAOS unit. During silent periods, while the call is still connected, the local TAOS unit will generate background (comfort) noise.
no	(Default) Silence is processed as part of the audio stream; comfort noise is not locally generated.

Changes to the parameter setting become effective with the next VoIP call.

The following example illustrates how to enable silence detection and comfort noise generation when processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set silence-det-cng = yes
admin> write
VOIP/{ 0 0 } written
```

When using silence suppression and comfort noise generation, the following apply:

- Silence compression and comfort noise generation must be enabled on both the local TAOS unit and distant TAOS unit involved in a call.

- When silence compression and comfort noise generation are enabled, the dynamic jitter buffer is not used (ena-adap-jitter-buffer=no).

- When either the G.723 or G.723-6.4kps codec is specified

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- Comfort noise generation can be enabled for processing at a rate of 5.3Kbps. With comfort noise enabled, the 5.3Kbps processing can decode silence detection and suppression packets.
- Comfort noise generation cannot be enabled for 5.3Kbps processing unless the adaptive jitter buffer is disabled.
- Silence detection/suppression cannot be enabled for 6.4Kbps processing unless the adaptive jitter buffer is disabled.

Adjusting the relative silence threshold

The silence-threshold parameter adjusts the relative threshold for silence suppression to compensate for background noise levels. This parameter lets the silence floor be raised or lowered in 1dB increments, without preventing conversation at normal speech levels from getting through. This parameter allows the user to raise the silence floor from an increase of 0 dB, the nominal level, to an increase of 9dB.

The following example illustrates how to raise the relative threshold for silence suppression for processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set silence-threshold = 3
admin> write
VOIP/{ 0 0 } written
```

Dependencies This parameter is ignored if silence suppression is disabled (silence-det-cng=no).

Configuring dynamic call jitter buffers

VoIP calls are processed using packet-based jitter buffering. A unique jitter buffer is opened for each call; the buffer dynamically adjusts its size to accommodate network jitter. In essence, jitter buffer playout delay adapts to network jitter.

To configure dynamic call jitter buffers, proceed as follows:

1   Enable the adaptive jitter buffer.
2   Configure the initial jitter buffer size.
3   Configure the maximum jitter buffer size.

Enabling the adaptive jitter buffer

The ena-adap-jitter-buffer parameter changes the jitter buffer mode to either adaptive or fixed for VoIP calls. When the adaptive mode is selected, the jitter buffer ranges in size between the values set for max-jitter-buffer-size and one packet, depending on the number of late or out-of-sequence packets received during the call. You can enter either of the following values:

Parameter value	Specifies
yes	(Default) That adaptive jitter buffering is used for processing VoIP calls.

Parameter value	Specifies
no	That static jitter buffers is used for processing VoIP calls.

Changes to this value become effective with the next VoIP call.

The following example illustrates how to enable or disable adaptive jitter buffering for VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set ena-adap-jitter-buffer = yes
admin> write
VOIP/{ 0 0 } written
```

Using adaptive jitter buffering has the following dependencies:

*   When silence-det-cng=yes, MultiVoice uses the value assigned to initial-jitter-buffer-size parameter to open static call jitter buffers.
*   When ena-adap-jitter-buffer=no, MultiVoice uses the value assigned to initial-jitter-buffer-size to open static call jitter buffers.
*   When G.723 codec is the valued for the packet-audio-mode parameter, the maximum jitter buffer size can't exceed nine packets (max-jitter-buffer-size=9).

Configuring the maximum jitter buffer size

The max-jitter-buffer-size parameter sets the maximum jitter buffer size for VoIP calls when the TAOS unit is configured to perform adaptive call jitter buffering. When using adaptive mode, the jitter buffer can increase to accommodate the entered number of audio packets, based on the incoming packet arrival statistics (jitter).

You may enter a value between 1 and 19 (packets). This allows the TAOS unit to expand the length of a call's jitter buffer to a size proportionate to the selected number of audio packets. This value defaults to 19. Changes to this value become effective with the next VoIP call.

The following example illustrates how to set the maximum jitter buffer size when adaptive jitter buffering is enabled for processing VoIP calls:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set max-jitter-buffer-size = 19
admin> write
VOIP/{ 0 0 } written
```

Configuring the initial jitter buffer size

The initial-jitter-buffer-size parameter sets the initial jitter buffer size for VoIP calls when the TAOS unit is configured to perform adaptive call jitter buffering. When using either adaptive or fixed mode, the jitter buffer is set to initial-jitter-buffer-size at start-up. During a call, the TAOS unit adjusts each jitter buffer to accommodate the number of audio packets, based on the in-coming audio packet volume.

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